An Acoustic Primer

Sharon Nieukirk and Anne Terhu
Oregon State University

What is sound?

Sound is a wave! More specifically a longitudinal wave.

→ The particles in a longitudinal wave move parallel to the direction in which the wave is traveling.

→ The particles in a transverse wave move perpendicular to the direction in which the wave is traveling (e.g., 'La Ola wave').

Sound wave

The medium through which the wave travels may experience some local oscillations as the wave passes, but the particles in the medium do **NOT** travel with the wave.

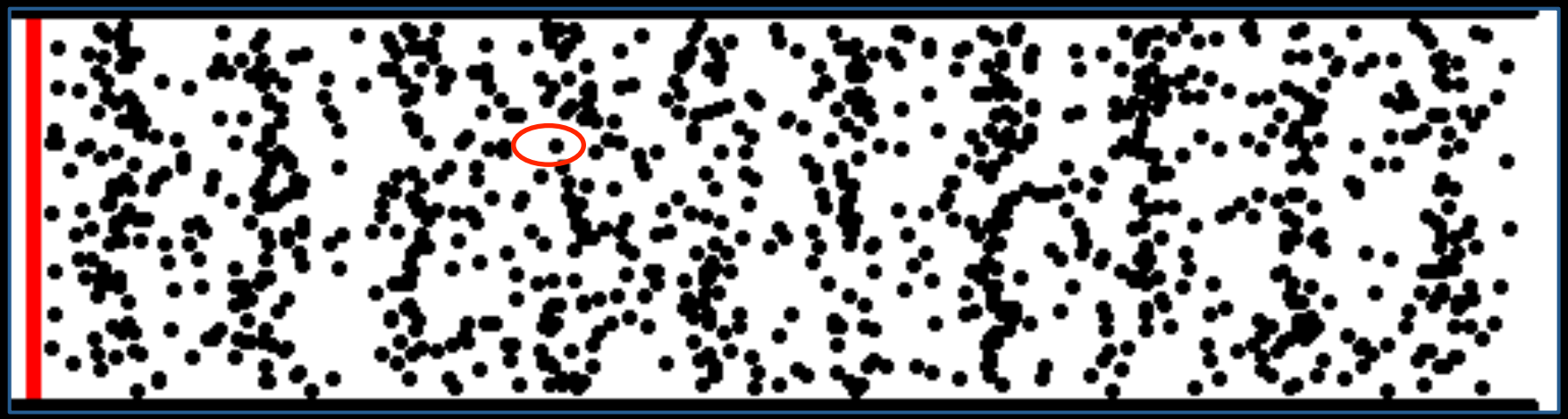
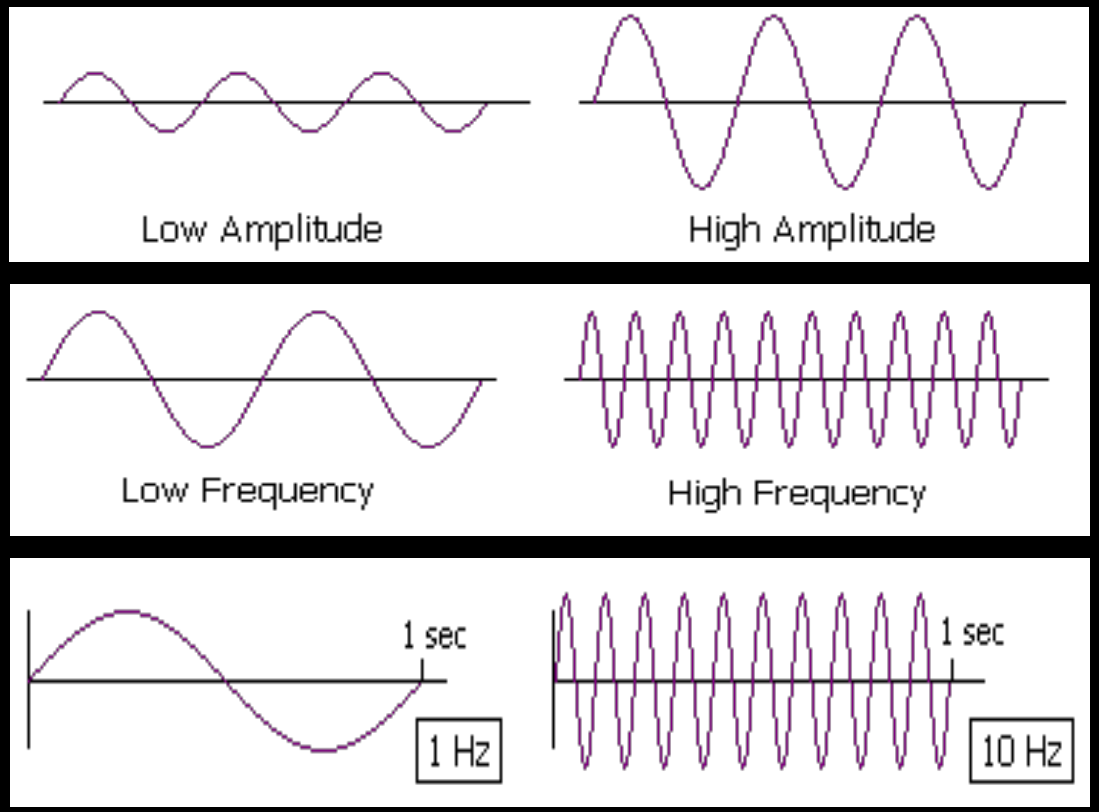


Image source: http://www.kettering.edu/physics/drussell/demos.html

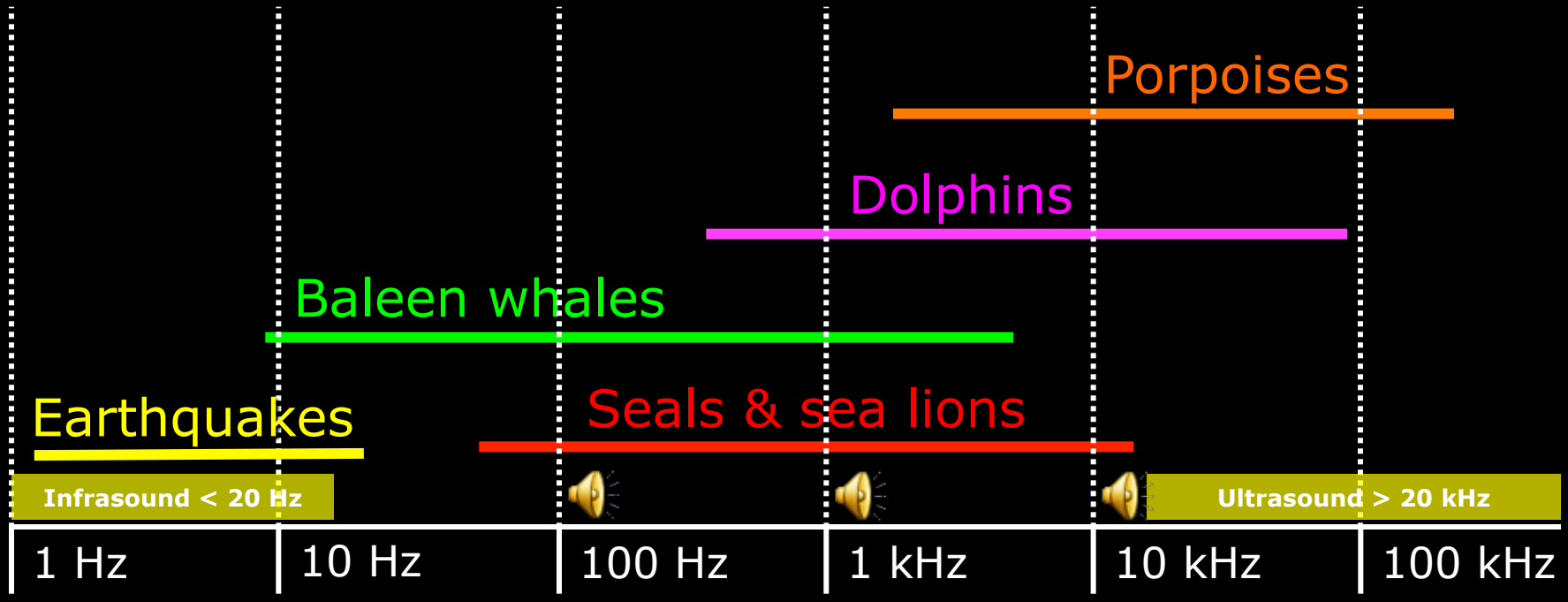
Measuring sound

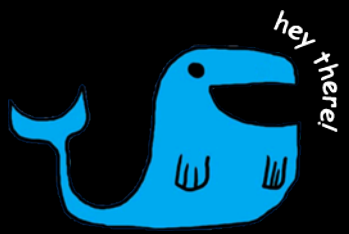
- Frequency [Hz]
- Wavelength [m]
- Amplitude [µPa]



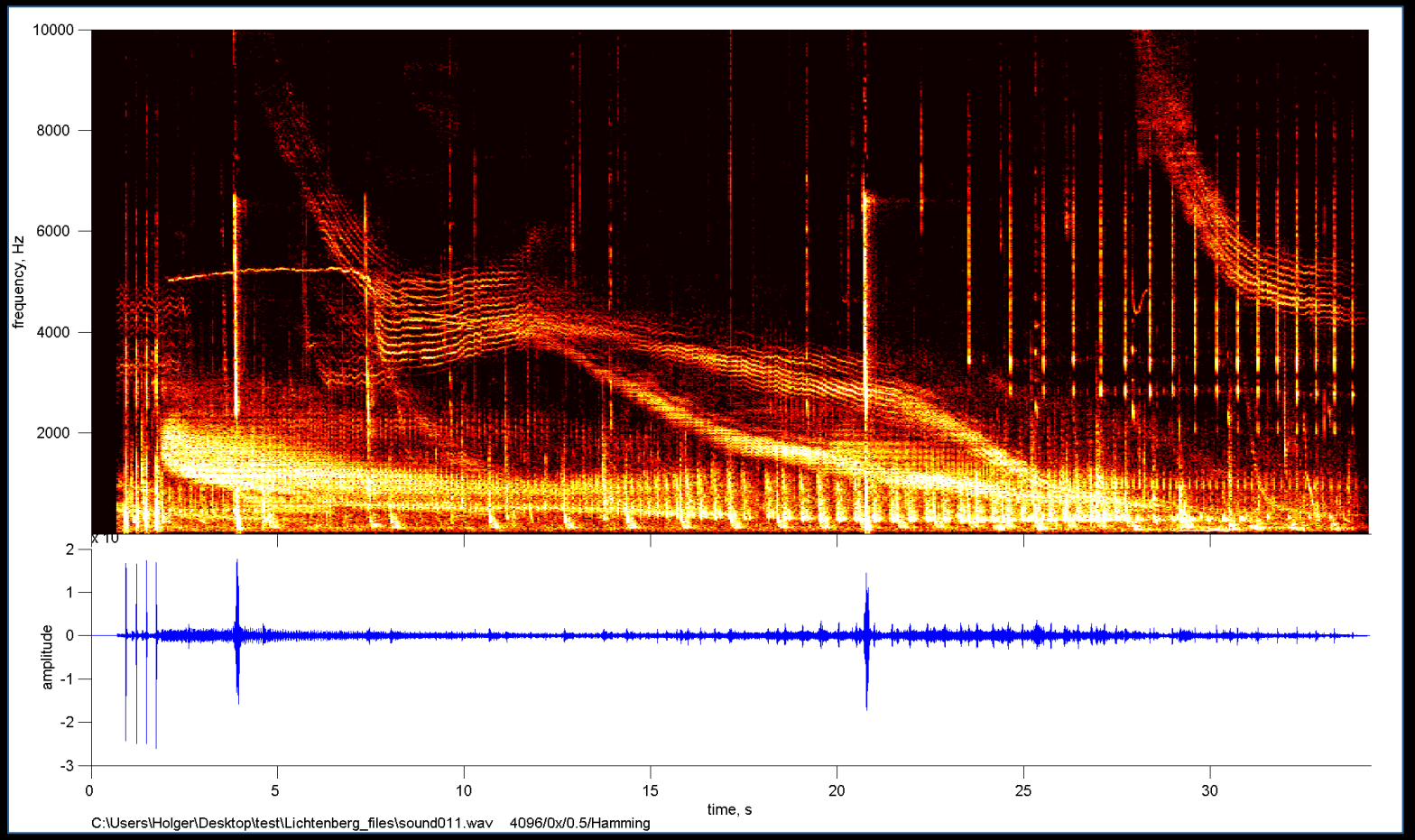
Great video on measuring sound (KHAN Academy):
https://www.khanacademy.org/science/physics/mechanical-waves-and-sound/sound-topic/v/sound-properties-amplitude-period-frequency-wavelength

Frequency


Porpoises
Dolphins
Baleen whales
Seals & sea lions
Earthquakes
Infrasound < 20 Hz
Ultrasound > 20 kHz
1 Hz
10 Hz
100 Hz
1 kHz
10 kHz
100 kHz
Frequency


hey there!

What is a spectrogram?



Sound in air/water

Sound in air and sound in water are both waves that move similarly and can be characterized the same way but…

…because liquids and gases have different properties such as density, they also have different sound speeds.

→ Sound travels ~1500 m/s in seawater (> 15 football fields end to end)
→vs. 340 m/s in air

→ The speed of sound in seawater is not a constant value. It changes with depth, temperature and salinity.

Sound speed in seawater

Sound speed is the most important environmental parameter that affects sound propagation in the ocean.

Sound Speed Equation:

$$c = 1449.2 + 4.6t - 0.055t^2 + 0.00029t^3 + (1.34-0.010t) * (s-35) + 0.0165z$$

c is sound speed in m/s
t is temperature in °C
s is salinity in ppt
z is depth in m

e.g., t = 20°C, s = 36 ppt, z = 10 m:
→ $c = 1449.2 + 4.6x20 - 0.055x20^2 + 0.00029x20^3 + (1.34-0.010x20) * (36-35) + 0.0165x10$
→ **c = 1522.825 m/s**

Decibels

The decibel (dB) is a logarithmic unit that indicates the ratio of a physical quantity relative to a specified or implied reference level.

→ Sound pressure level, **SPL = 20* $\log_{10}$ (p_s/p_r)**

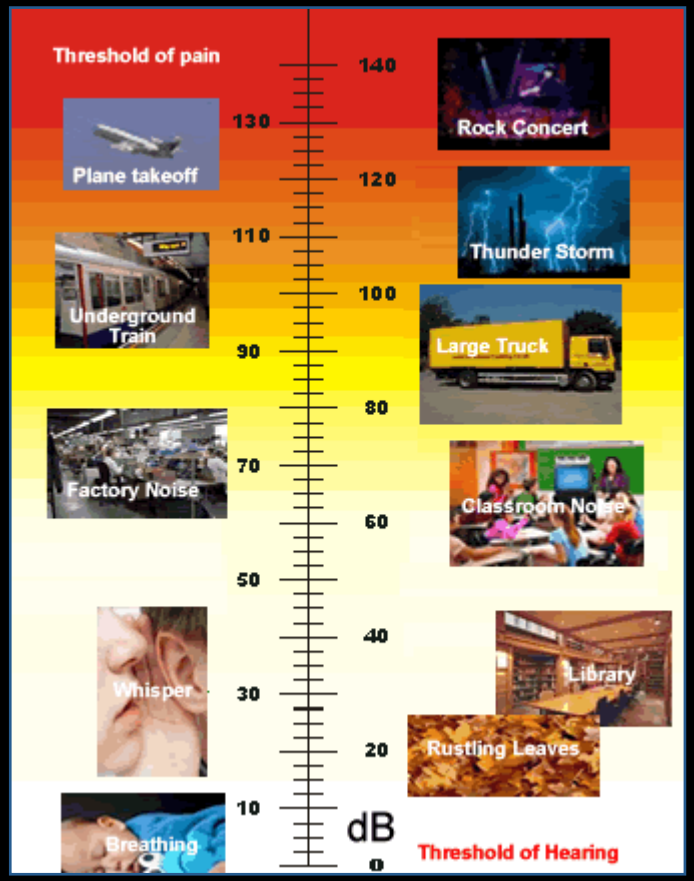
p_s = pressure of an acoustic signal **[µPa]**
p_r = reference pressure; **in water 1 µPa**
SPL = sound pressure level **[dB re 1 µPa]**

→ 2x p_s will result in +6 dB
→ 10x p_s will result in +20 dB
→ 100x p_s will result in +40 dB

Don’t compare apples and oranges!

dB in air and water are not the same!

AIR:



Source: http://tsalveta.info/ebook/hearing.html

Characterizing sounds

dB in air and water is not the same!

Caused by different:

[a] reference pressures

- in water 1 µPa
- in air 20 µPa

[b] densities and sound speeds

→Intensity measurements of equal pressures in air and water differ by ~ 62 dB!

recreated from DOSITS.org

Don’t compare apples and oranges!

An example (source level, SL):

Gray whale moans: 142 - 185 dB re 1 µPa @ 1 m

→ In air: 80 - 123 dB re 20 µPa @ 1 m

In comparison:
- Lawn mower: ~90 dB re 20 µPa @ 1 m
- Auto horn: ~110 dB re 20 µPa @ 1 m

ALWAYS provide information on the reference pressure!

Similar problem: degree temperature (°F or °C ???)

Source: http://cetus.ucsd.edu/voicesinthesea_org/Flash/

Don’t compare apples and oranges!

Action Items!

Our friends at the Center for Biological Diversity have alerted us to an immediate, tangible action that we all can, and should do. By July 14.

We are looking at seismic surveys in the Gulf of Mexico.
This issue is near and dear to us as it involves whales and dolphins. It also addresses sound in marine settings, which is one of the things that our next film will be looking at.
Oil and gas companies use seismic surveys to find underwater sources, but the process involves blasts of high-powered air guns, which can exceed 250 decibels. (A vacuum cleaner measures around 80 dB, front row at a rock concert can be 110 dB, a jet at takeoff averages 120-140 dB).
Naturally this impacts all ocean life, but of specific concern is the hearing loss in marine mammals. Remember, these magnificent creatures rely on sound for food, mating, and survival.
This application is now before the National Marine Fisheries Service for review. Scroll for the specific item to read and understand. These activities are perceived as harrasment and are in violation of the Marine Mammal Protection Act.
There is a period of time where it is open to public comment. That means we must do one (or all) of the following.
Go here and add your name.
http://action.biologicaldiversity.org/p/dia/action/public/?action_KEY=7163

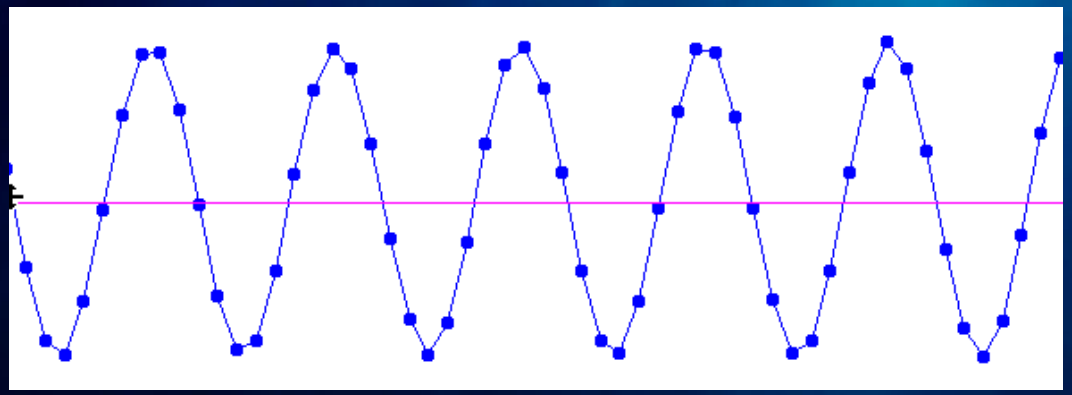
You can also write a brief, respectful letter to the address below sharing your thoughts or ask for further information. This must be received by July 14.
P. Michael Payne, Chief
Permits, Conservation, and Education Division
Office of Protected Resources, NMFS
1315 East-West Highway
Silver Spring, MD 20910-3225

Thanks all.

Source: http://email.maildiva.com/t/ViewEmail/r/CC5B6185852F6292/D54AE5B66E0381F0C68C6A341B5D209E

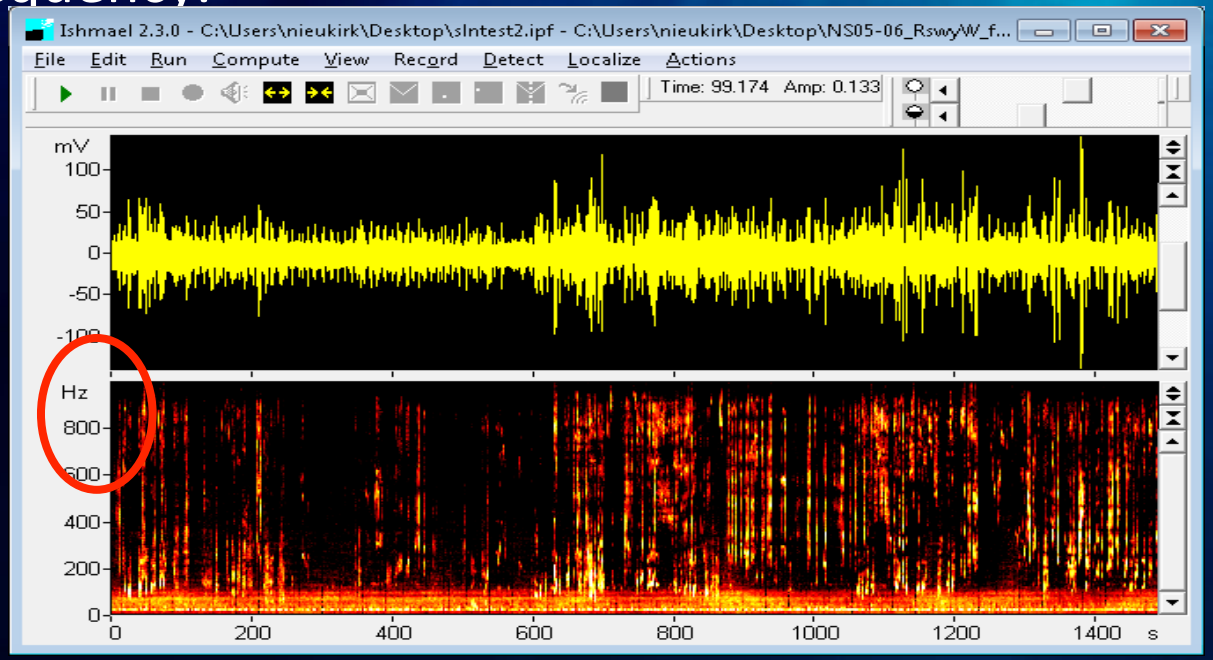
A word about sample rate……

- Computers and DAT recorders sample (digitize) the continuous rise and fall of sound amplitudes at some fixed rate and store a long column (vector) of amplitude values. Music CDs sample at 44.1 kHz (or 44,100 samples/s).



Sample rate - Nyquist frequency

- **Nyquist frequency**: A digital recorder or computer must be able to take at least 2 samples/cycle to be able to identify each frequency.
- Thus, if you digitize your sounds at *R* samples/sec, you will be unable to properly capture any component with frequency >*R*/2. This latter value is called the Nyquist frequency.



Nyquist = 1000 Hz

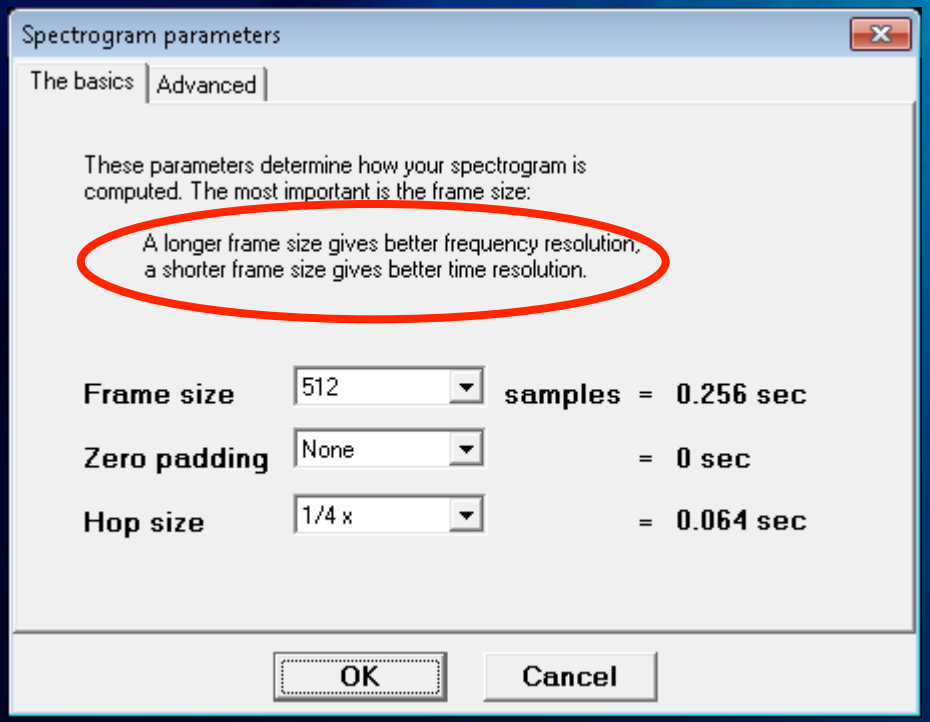
sr = 2000 Hz

Adapted from
Bradbury & Vehrencamp
Cornell University

Ishmael: a word about spectrogram settings……

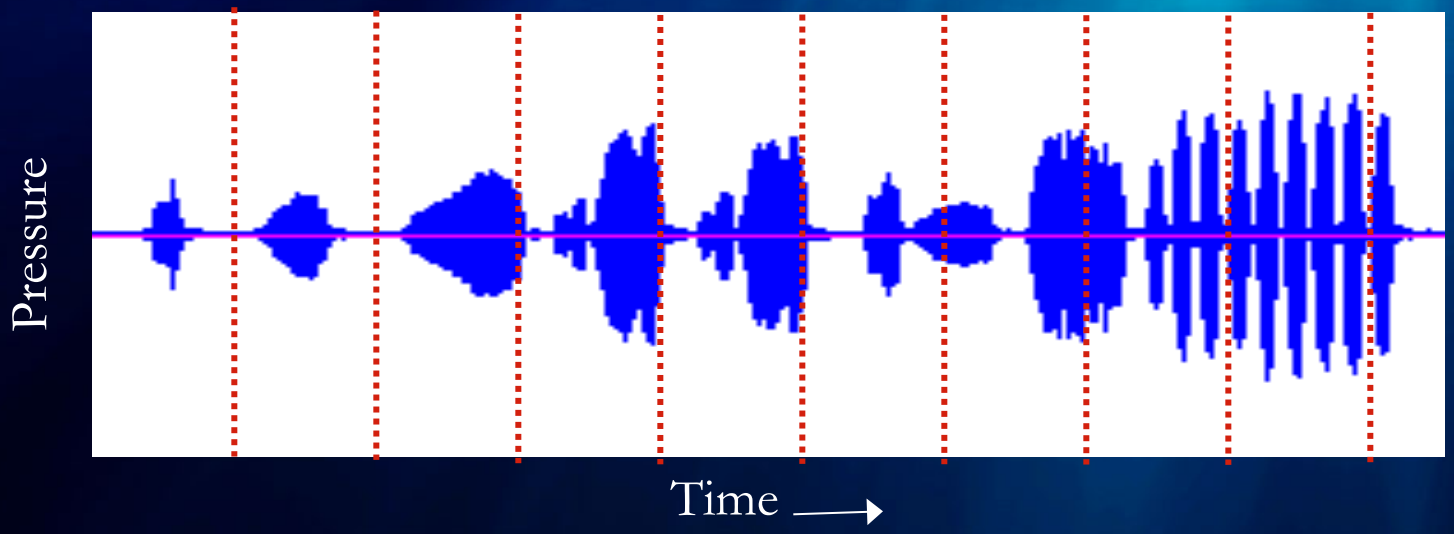
Exercise 3:

- Launch Ishmael
- Open a sound file (NS04-05_RswyW_file110.wav)
- Change spectrogram settings (*Compute----Spectrogram parameters*)



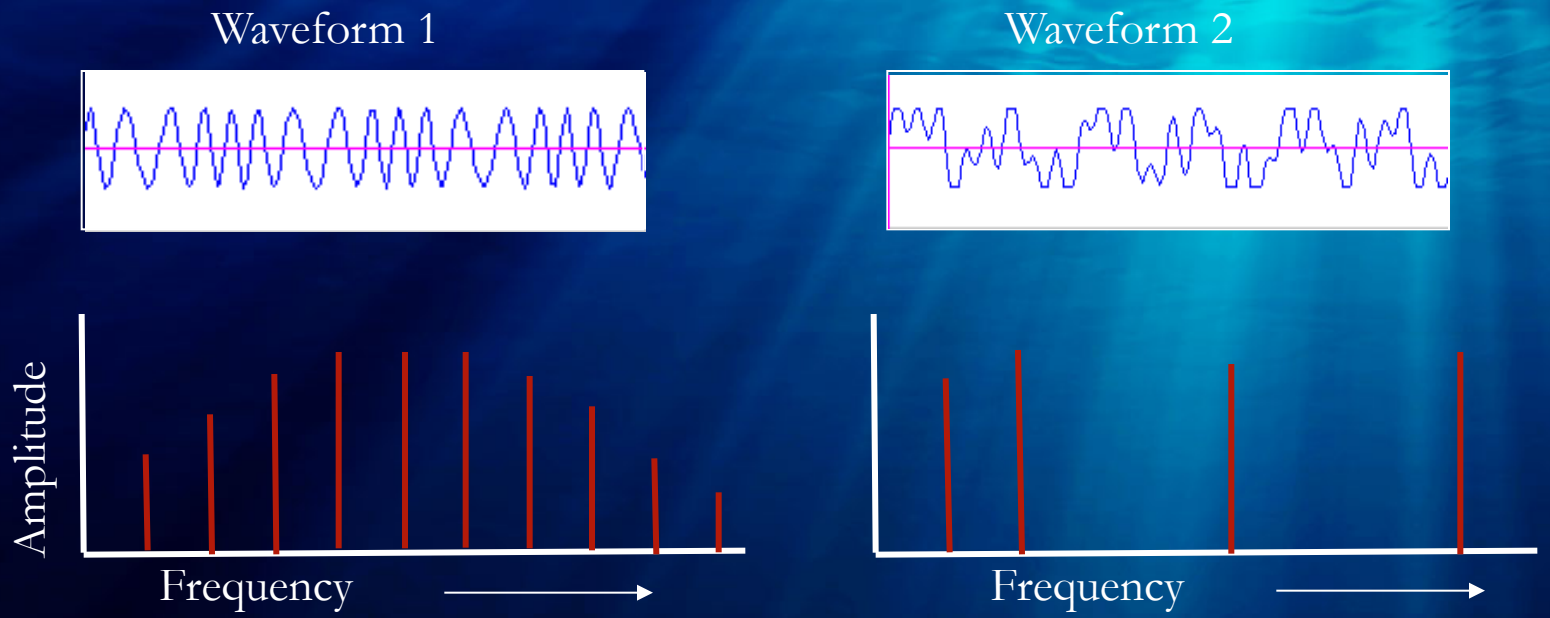
Fourier Analysis

- Waveform keeps changing during the signal
- Break the song into homogeneous segments and create a frequency spectrum for each segment.



Fourier Analysis

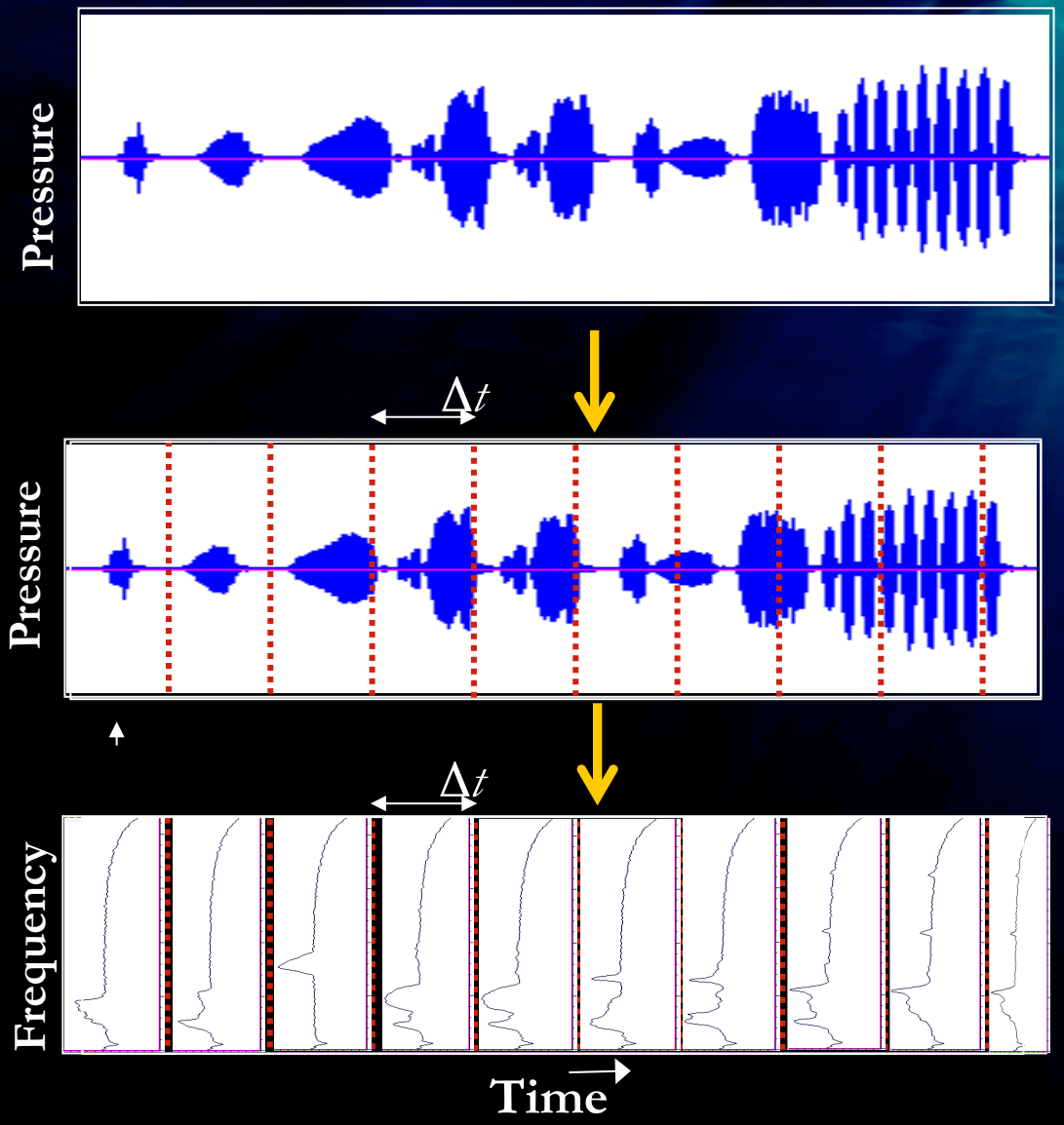
- Applying the Fourier algorithm, we get:



✓A plot of amplitude versus frequency components is called the *frequency spectrum* (or *power spectrum*) of a sound.

Bradbury & Vehrencamp
Cornell University

Creating a spectrogram



Time domain

FFT

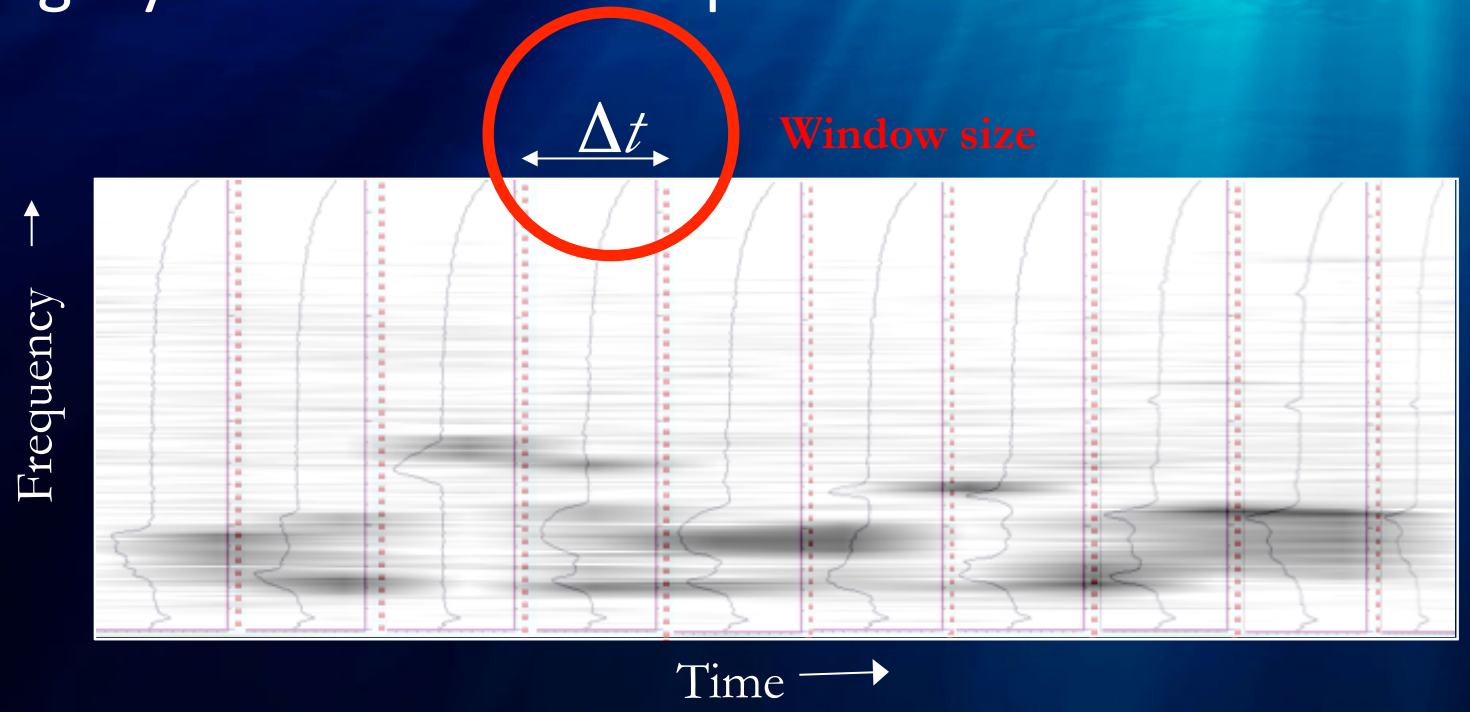
- dividing a sound into segments
- computing the frequency spectrum for each segment
- stringing the segments together along the time axis

Frequency domain

Adapted from
Bradbury & Vehrencamp

Creating a spectrogram

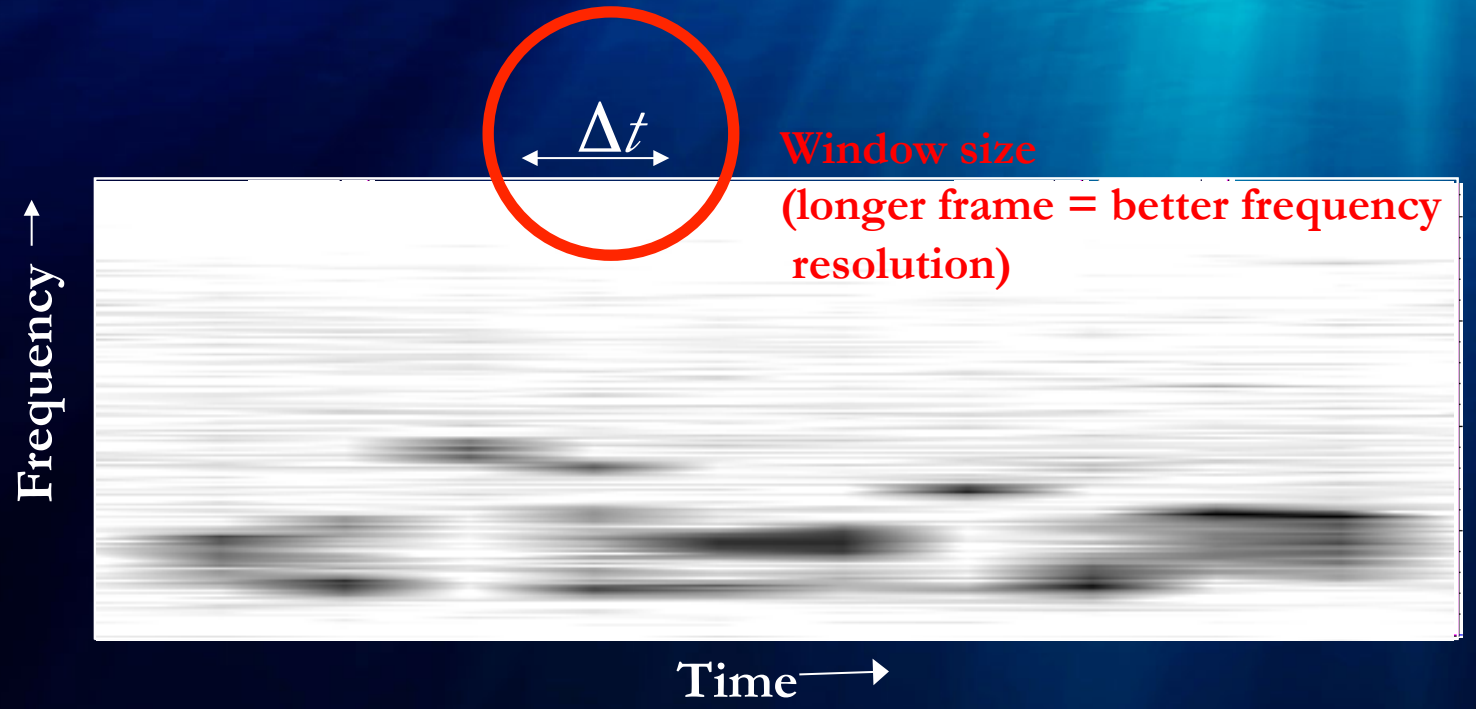
- Then, use black to mark those portions of the overall graph that have higher peaks, use white to mark the lower amplitude components, and use grey for intermediate portions.



Adapted from
Bradbury & Vehrencamp

Creating a spectrogram

- The result is a spectrogram with frequency on the vertical axis, time on the horizontal axis, and amplitude of a frequency component at a given time indicated by darkness on the plot.

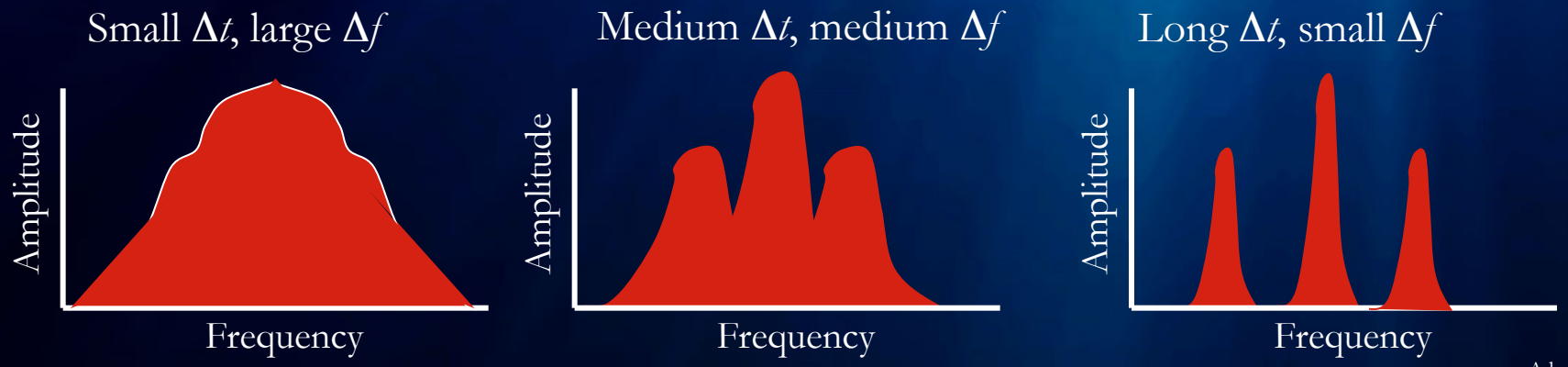


Adapted from
Bradbury & Vehrencamp

The Uncertainty Principle

- If we let Δt be the duration of the shortest sampling time available to a Fourier analyzer, the **Uncertainty Principle** for sound analysis states that:

$$\Delta f \cdot \Delta t \approx 1$$ (long window = good freq resolution)



Adapted from
Bradbury & Vehrencamp

The Uncertainty Principle

Frequency resolution and time resolution are inversely related…

$$\Delta f \cdot \Delta t \approx 1$$

- Long window (Δt) = good frequency resolution
- Short window (Δt) = good time resolution

Decide on your question (and signal of interest) then choose your spectrogram settings!

Sample rate also matters….

Sample rate limits your spectrogram settings

Sample rate / window size = frequency resolution

Example: 44,100 samples/s / 1024 samples = 43 Hz res.

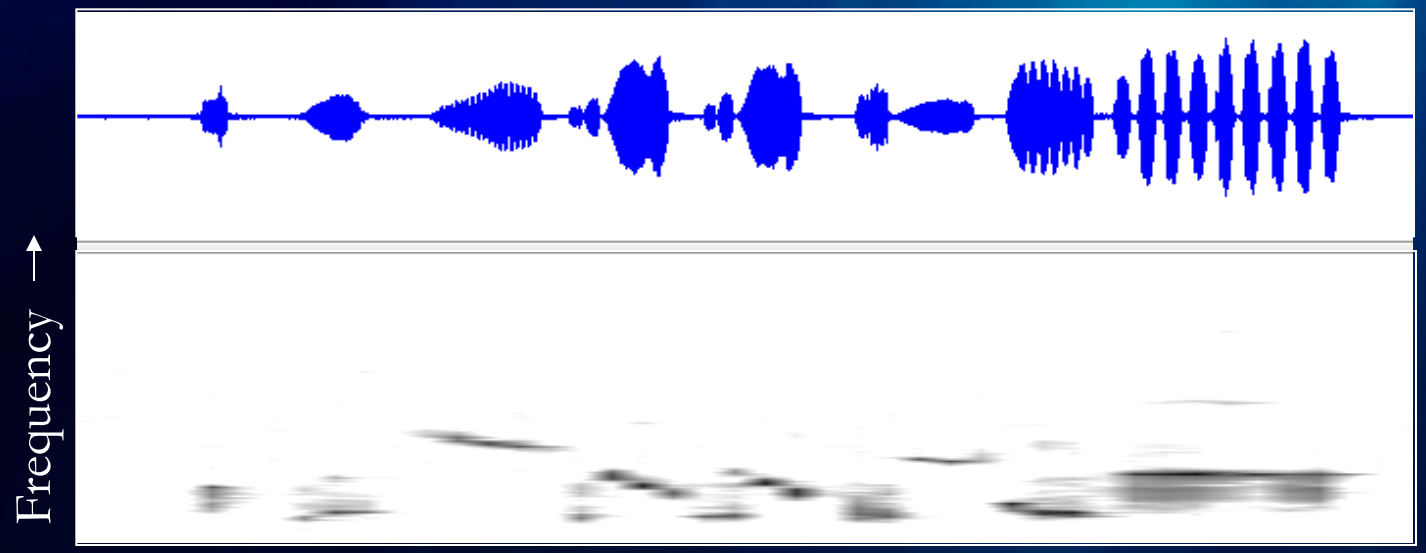
Spectrograms and Bandwidth

- **The spectrogram we just made uses a pretty large Δt. This gives us very fine frequency resolution (Δf = 5 Hz), but much of the temporal resolution has been lost. Can we get by with a smaller Δt?**

Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

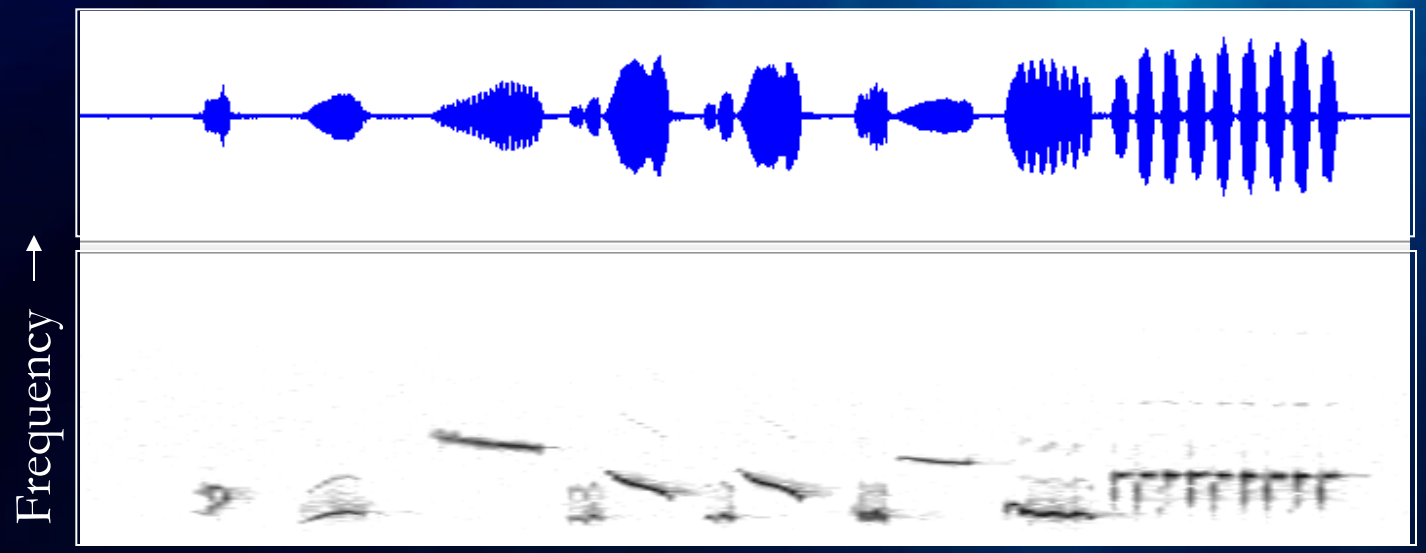
- Let's decrease Δt by 4×. This will give us a Δf = 20 Hz). This starts to restore some of the temporal pattern, and the frequency bands are still pretty thin.



Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

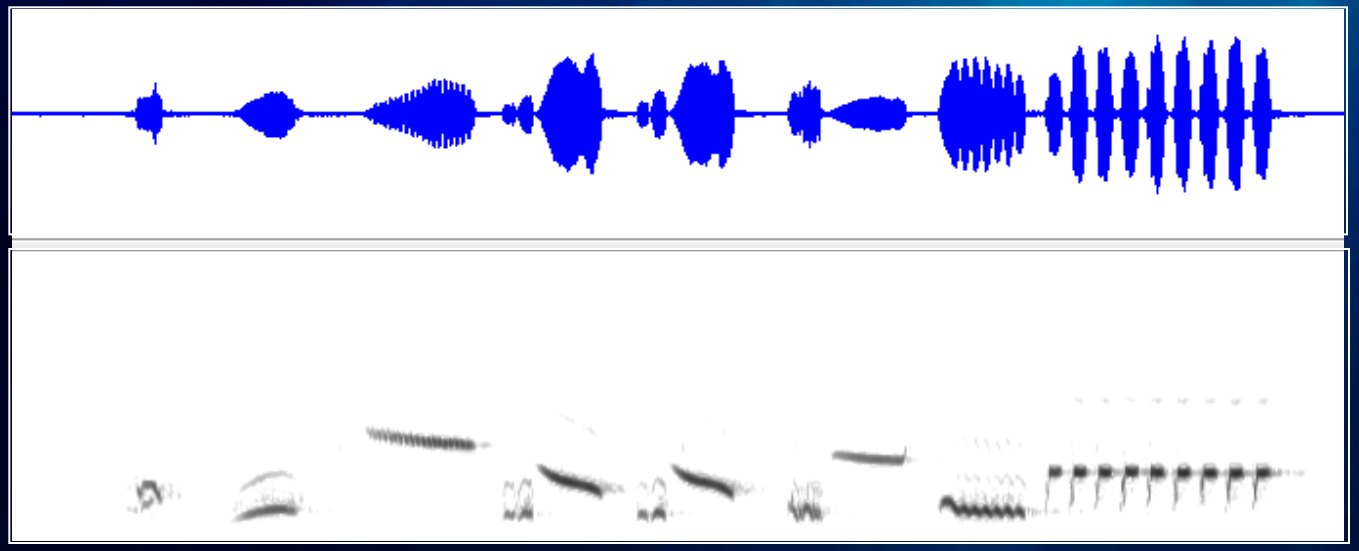
- Let's decrease Δt by 4× again. This will give us a Δf = 80 Hz. We get much better temporal pattern and even some better frequency pattern because FM signals show as FM, not their components!



Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

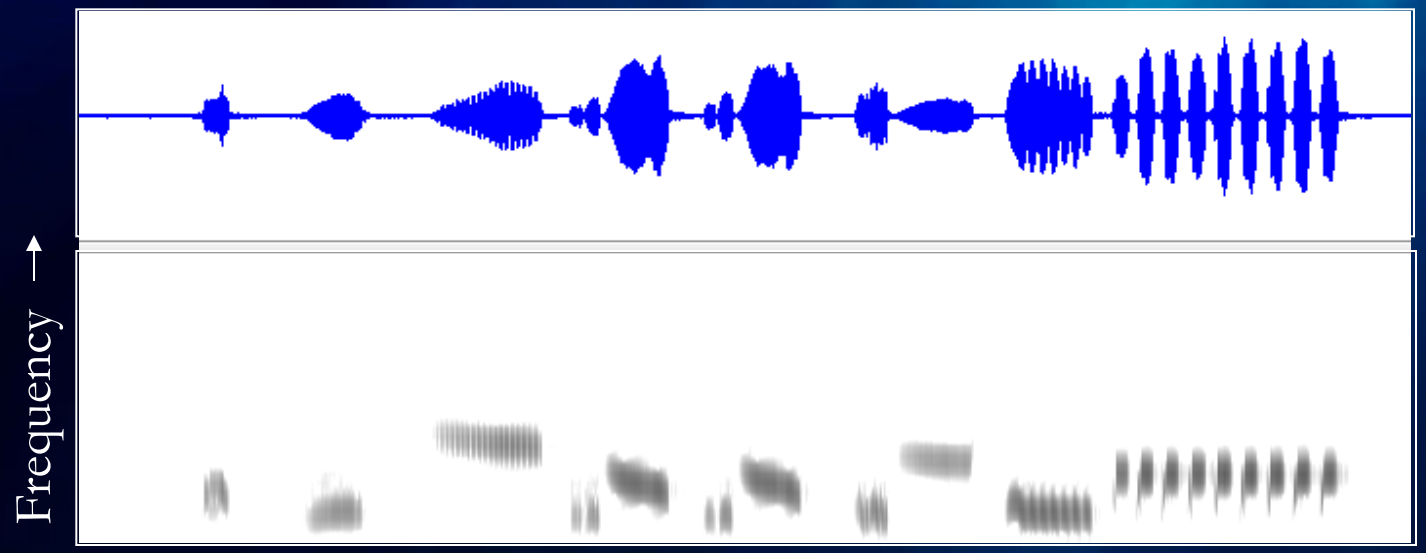
- Let's decrease Δt by 4× once more. This will give us a Δf = 320 Hz. This is similar to the prior bandwidth, but we can see the temporal pattern in the last notes better.



Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

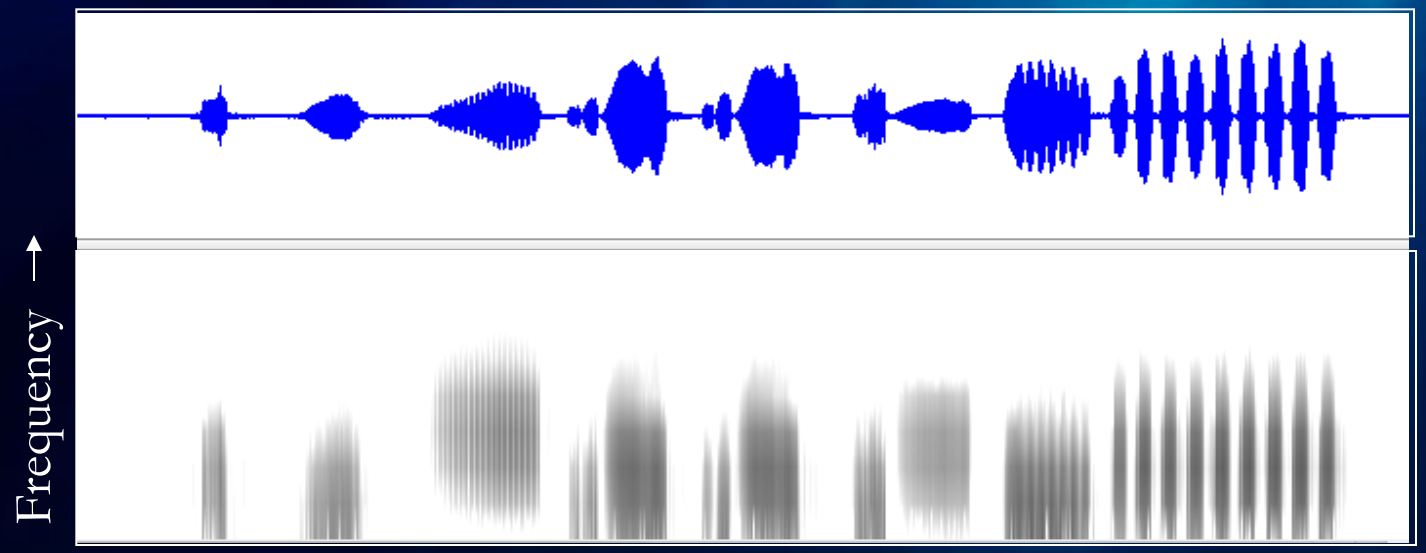
- Let's decrease Δt by 4× again. This will give us a Δf = 1280 Hz. Now, large bands start to appear instead of fine lines, although the temporal pattern is retained.



Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

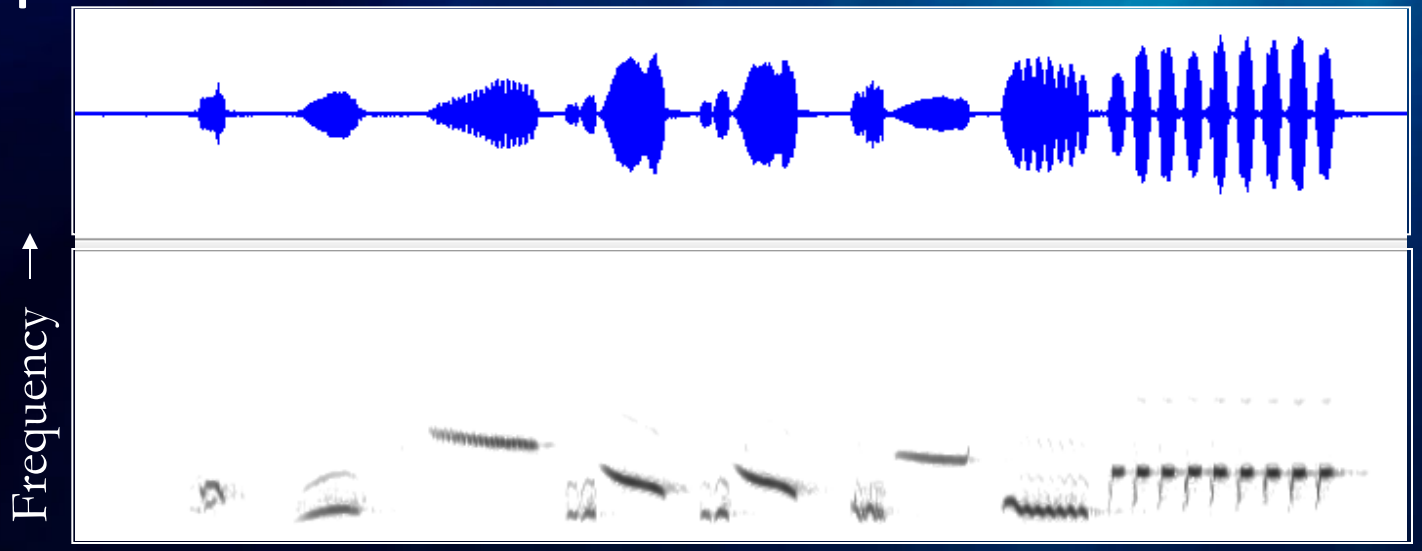
- Let's decrease Δt by 4× yet again. This will give us a Δf = 5120 Hz. We have now lost any decent frequency resolution, but the temporal pattern is retained.



Adapted from
Bradbury & Vehrencamp

Spectrograms and Bandwidth

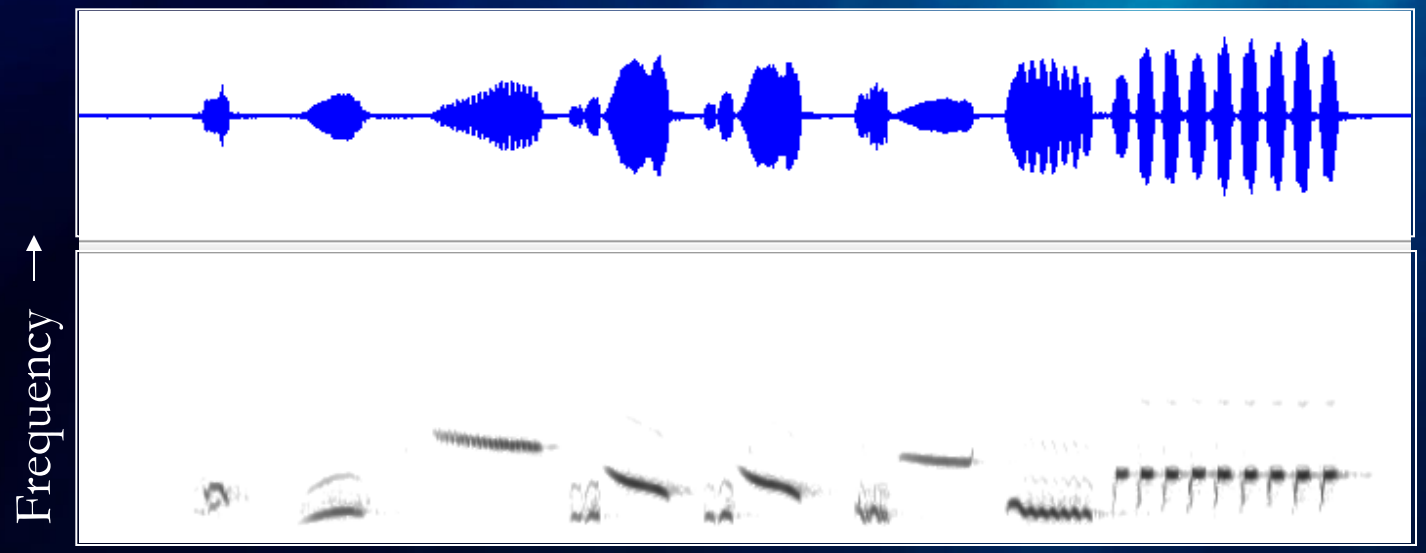
- An intermediate bandwidth, Δf, provides the optimal balance of frequency resolution and temporal resolution.
- Choose your settings based on your question!



Adapted from
Bradbury & Vehrencamp

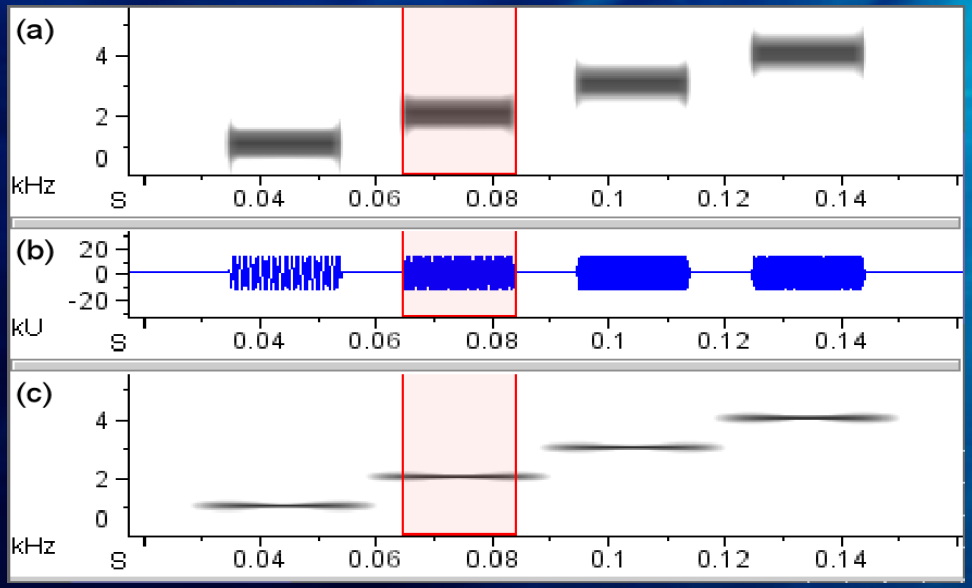
Spectrograms and Bandwidth

- In general, you want a bandwidth:
  - ✓small enough to separate harmonics clearly;
  - ✓big enough to show FM undecomposed; and
  - ✓big enough to show AM undecomposed.



Adapted from
Bradbury & Vehrencamp

Spectrogram parameters



Short window - good time resolution
64 points (2.90 mS)

long window - good freq. resolution
512 points (23.2 mS)

Effect of record length and filter bandwidth on time and frequency resolution. The signal consists of a sequence of four tones with frequencies of 1, 2, 3, and 4 kHz, at a sampling rate of 22.05 kHz. Each tone is 20 mS in duration. The interval between tones is 10 mS. Both spectrograms have the same time grid spacing = 1.45 mS, and window function = Hann. The selection boundaries show the start and end of the second tone.

(a) Wide-band spectrogram: record length = 64 points (= 2.90 mS), 3 dB bandwidth = 496 Hz.

(b) Waveform, showing timing of the tones.

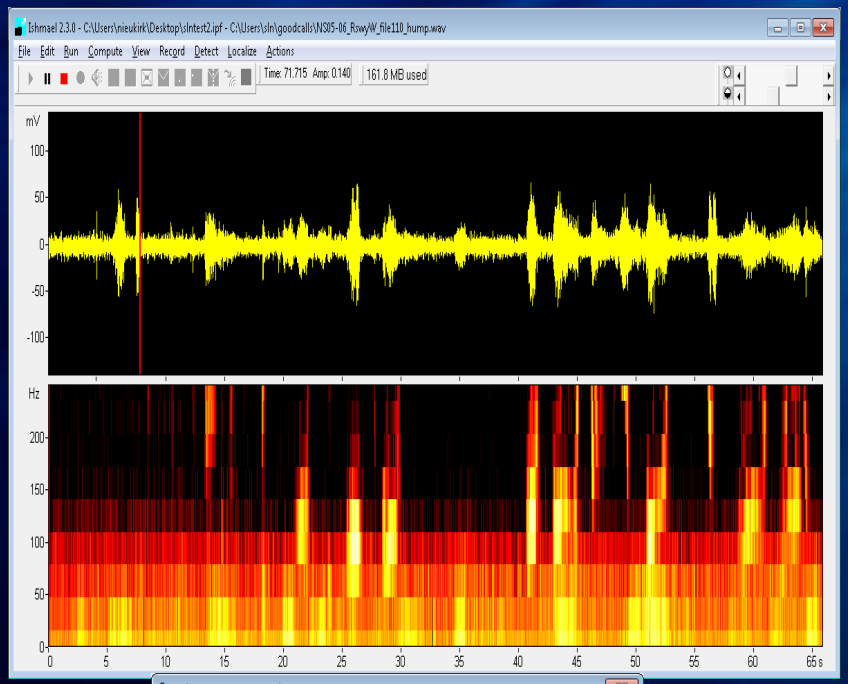
(c) Narrow-band spectrogram: record length = 512 points (= 23.2 mS), 3 dB bandwidth = 61.9 Hz.

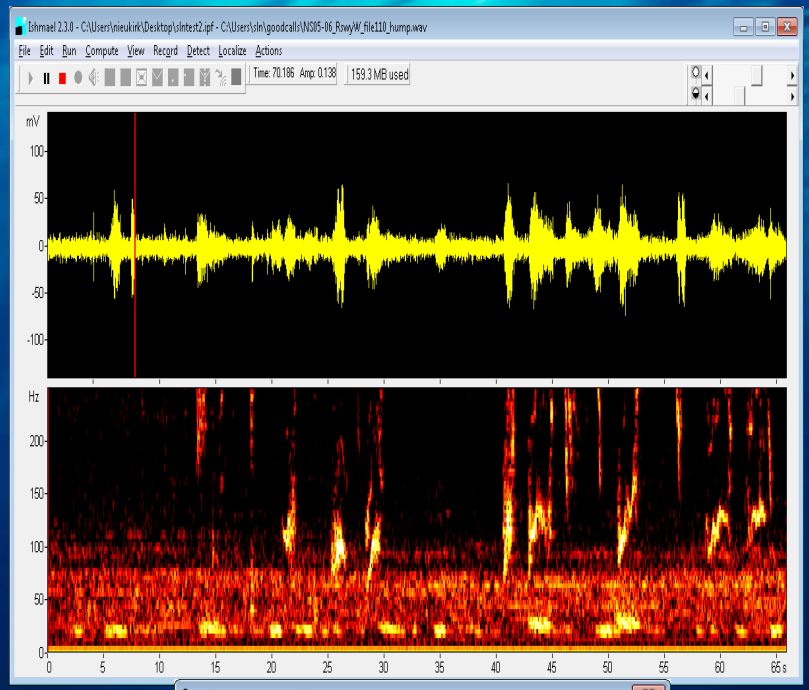
The waveform between the spectrograms shows the timing of the pulses.

Adapted from Raven User's Manual

Ishmael

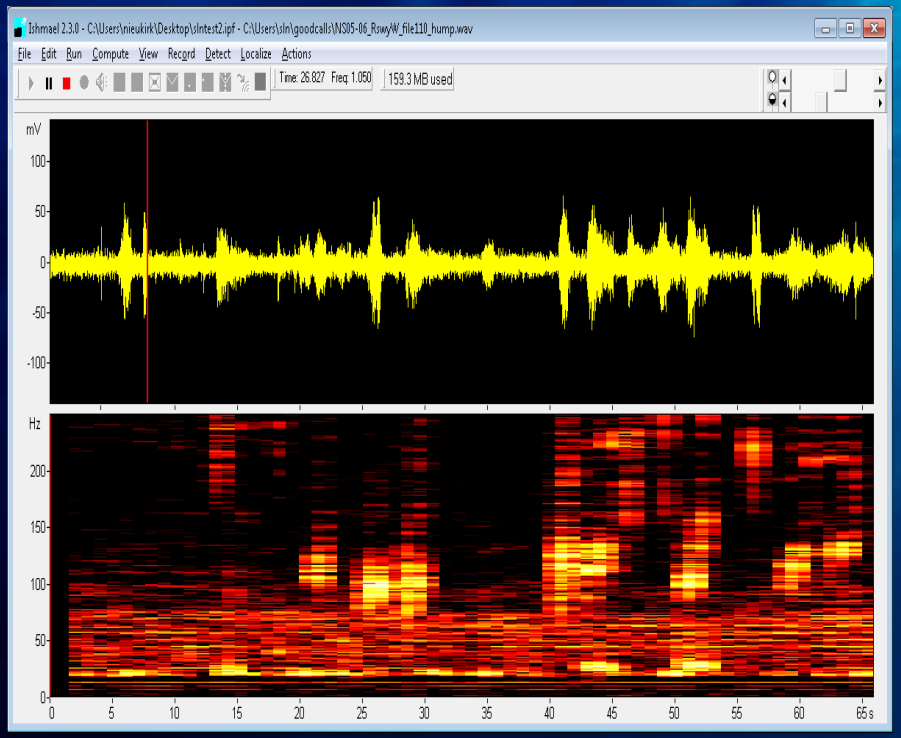
Exercise 3: Change spectrogram settings

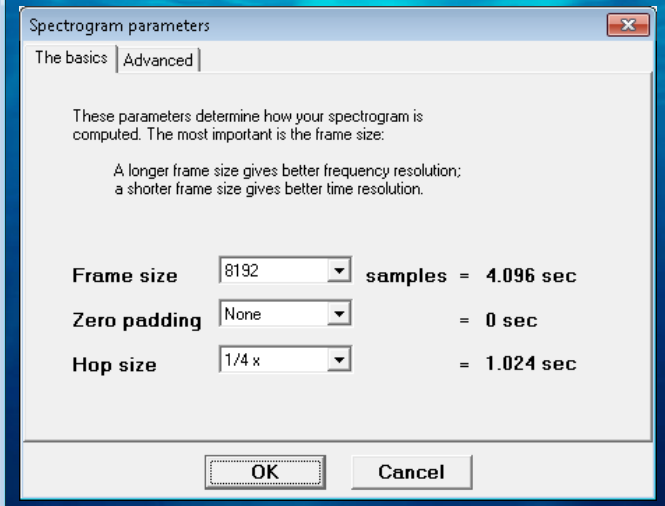




Ishmael

Exercise 3: Change spectrogram settings – there is a limit…….



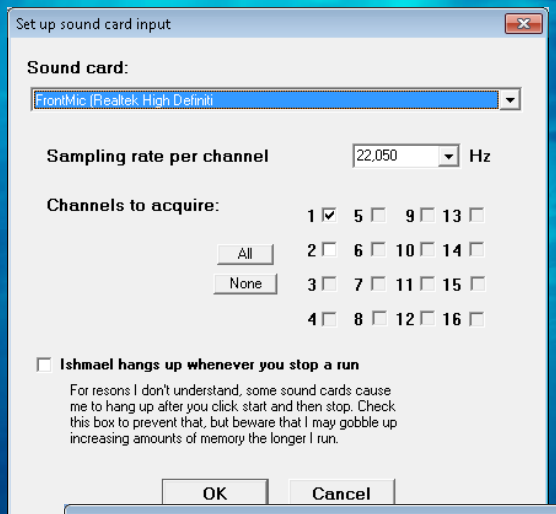


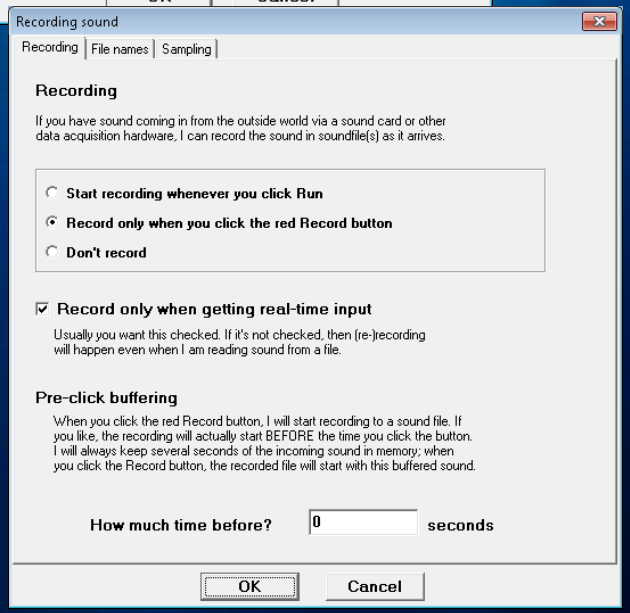
Conclusion: choose your settings based on your question!

Ishmael: real-time data acquisition

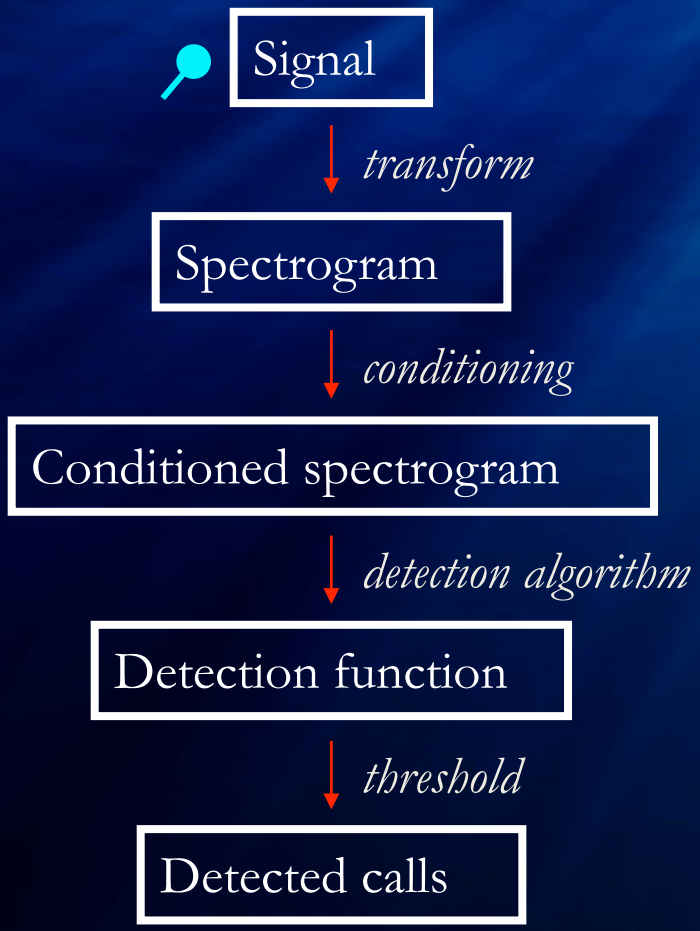
To view and record real-time data

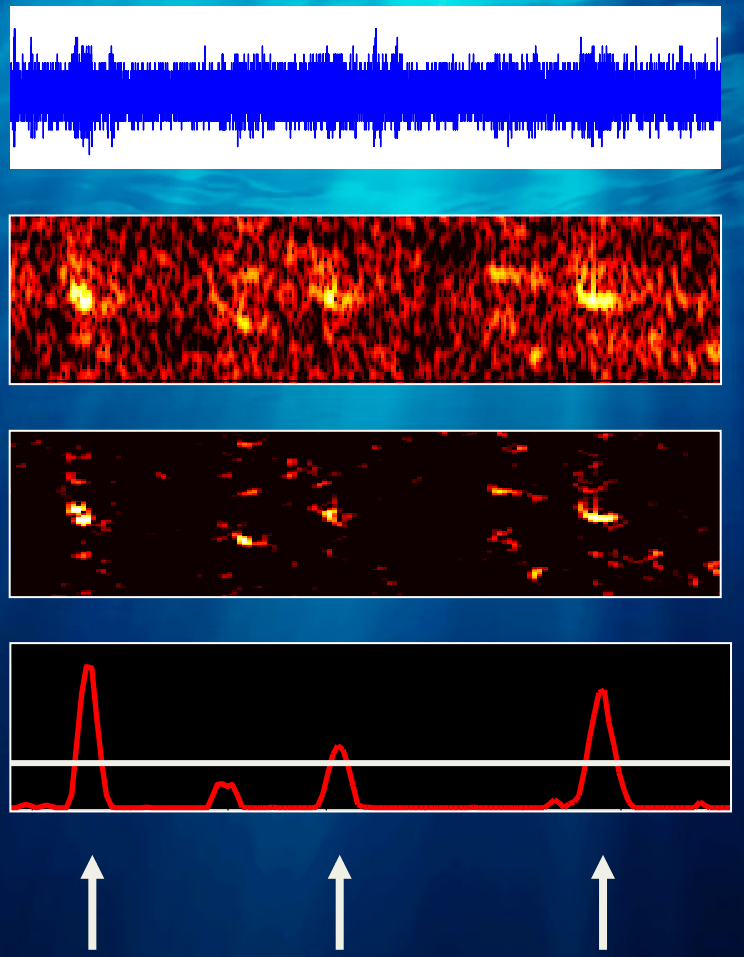
- Plug microphone in FIRST
- Launch Ishmael
- Under File menu, check soundcard setting
- Under Record menu, choose “record only when you click the red button” and “record only when getting real-time input”


Set up sound card input
Sound card:
FrontMic (Realtek High Definiti
Sampling rate per channel
22,050
Hz
Channels to acquire:
All
None
1 5 9 13
2 6 10 14
3 7 11 15
4 8 12 16
Ishmael hangs up whenever you stop a run
For resons I don't understand, some sound cards cause me to hang up after you click start and then stop. Check this box to prevent that, but beware that I may gobble up increasing amounts of memory the longer I run.
OK
Cancel


Recording sound
Recording
File names
Sampling
Recording
If you have sound coming in from the outside world via a sound card or other data acquisition hardware, I can record the sound in soundfile(s) as it arrives.
Start recording whenever you click Run
Record only when you click the red Record button
Don't record
Record only when getting real-time input
Usually you want this checked. If it's not checked, then (re-)recording will happen even when I am reading sound from a file.
Pre-click buffering
When you click the red Record button, I will start recording to a sound file. If you like, the recording will actually start BEFORE the time you click the button. I will always keep several seconds of the incoming sound in memory; when you click the Record button, the recorded file will start with this buffered sound.
How much time before?
0
seconds
OK
Cancel

Ishmael: automatic detection

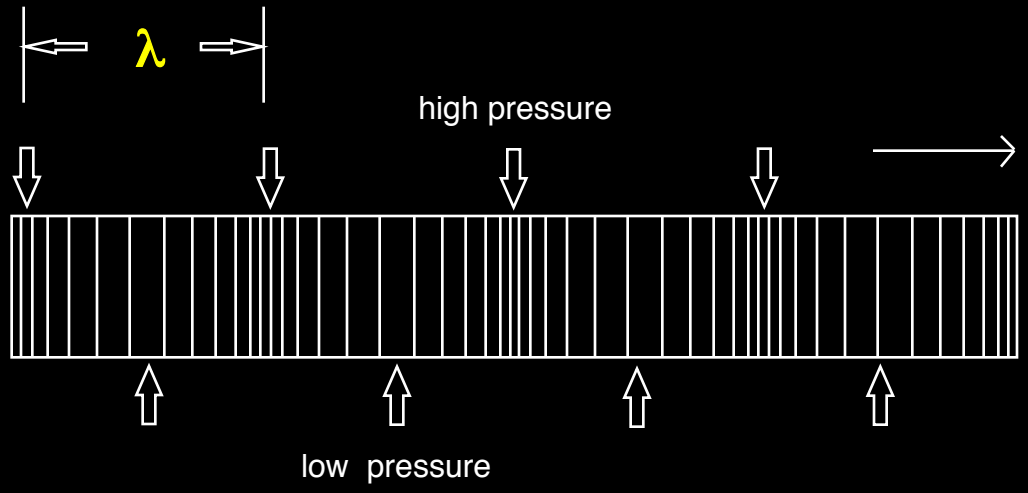




Sound wave metrics

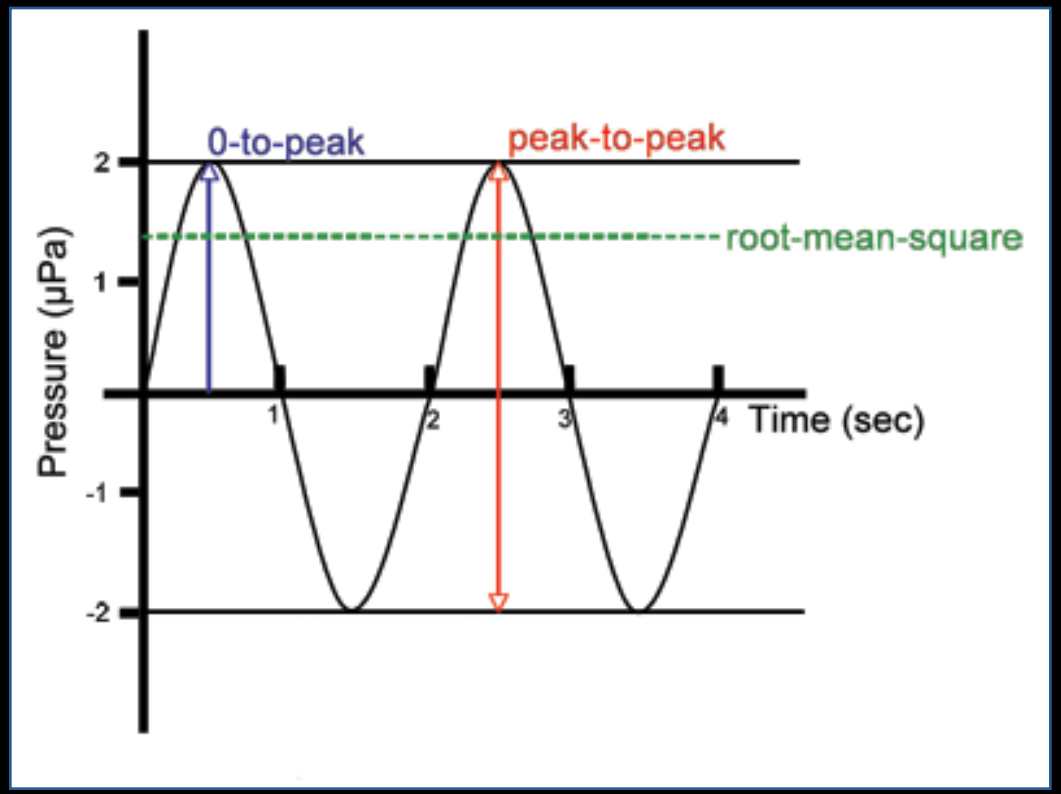
Frequency f and wavelength λ

$\lambda = c/f$ with c is sound speed in m/s and f frequency in Hz



Measuring sound

- Frequency [Hz]
- Wavelength [m]
- Amplitude [µPa]



Source: http://www.dosits.org/science/advancedtopics/signallevels/